User Manual

BudgeTone-100 Series

IP Phone

For Firmware Version 1.0.8.16

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1 Welcome

Congratulations on becoming an owner of BudgeTone-100 IP telephone! You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's award-wining BudgeTone-100 series of SIP phones are innovative IP telephones that offer a rich set of functionality and superb sound quality at ultra-affordable price. They are fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Grandstream BudgeTone-100 IP telephone has been awarded the Best of Show product in 2003 Internet Telephony Conference and Expo.

This document is subject to changes without notice. The latest electronic version of this user manual can be downloaded from Grandstream Networks' official website:

http://www.grandstream.com/user_manuals/budgetone100.pdf

2 Installation

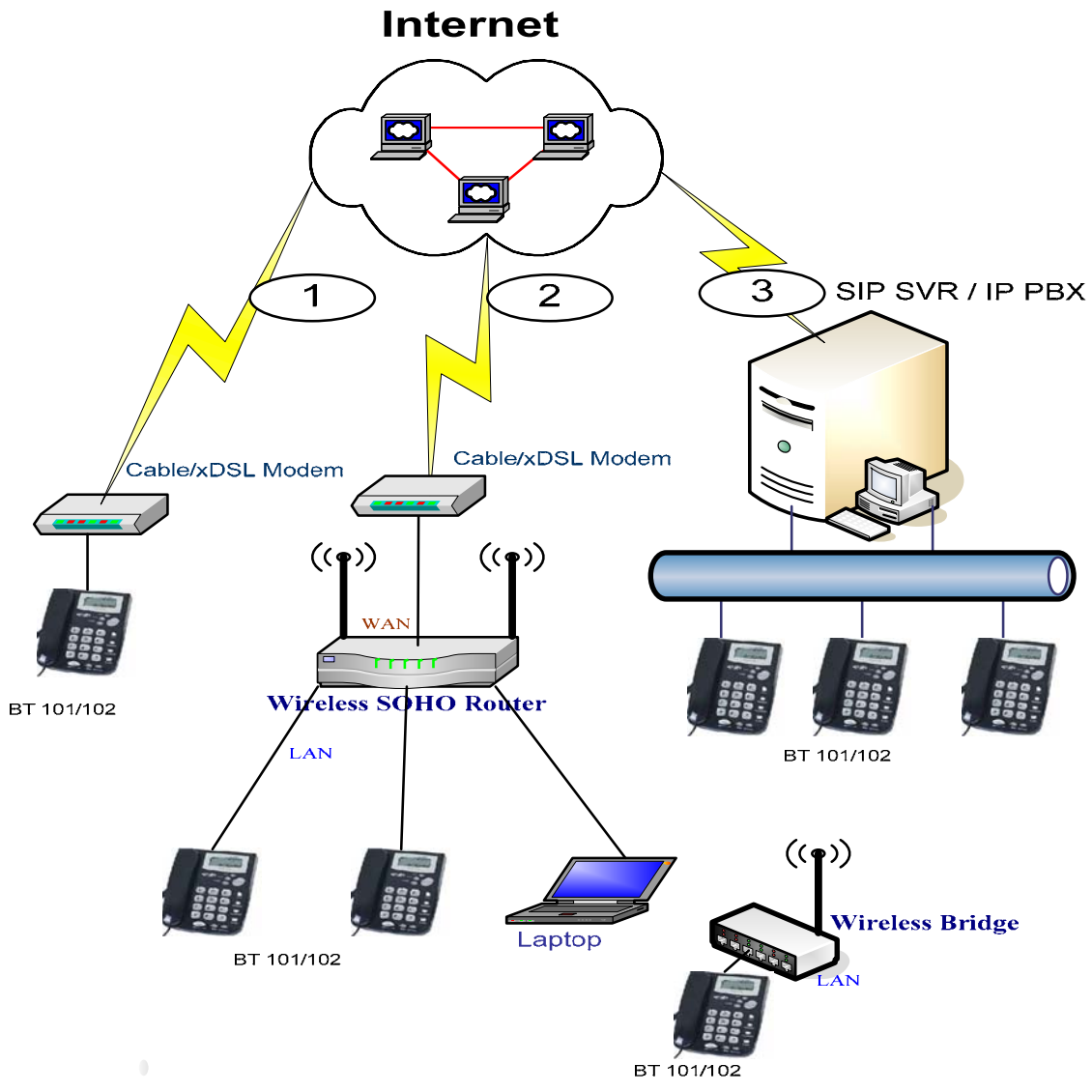
BudgeTone-100 series IP phones are designed to look and feel like standard telephones. The following photo illustrates the appearance of a BudgeTone IP phone and the use of its key buttons.

2.1 Interconnection Diagram

There are several ways to connect the BT100 series IP telephone:

1. Connected directly behind Cable/DSL modem
2. Connected to LAN side of a (wireless) SOHO router *(This is the most popular connection)*.
3. Connected to an Ethernet network

Following diagram illustrates the interconnection of phones in above-mentioned networks:



3 What is Included in the Package

The BudgeTone-100 phone package contains:

1) One BudgeTone-100 phone
2) One universal power adaptor
3) One Ethernet cable

3.1 Safety Compliances

The BudgeTone-100 phone is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. The phone should only operate with the universal power adaptor provided with the package. Damages to the phone caused by using other unsupported power adaptors would not be covered by the manufacturer's warranty.

3.2 Warranty

Grandstream has a reseller agreement with our reseller customer. End user should contact the company from whom you purchased the product for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number.

Grandstream reserves the right to remedy warranty policy without prior notification.

Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the BudgeTone-100 IP telephone and will void the manufacturer warranty.

Caution: Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

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4 Product Overview

Grandstream IP Phone is a next generation IP network telephone based on industry open standard SIP (Session Initiation Protocol). Built on innovative technology, Grandstream IP Phone features market leading superb sound quality and rich functionalities at mass-affordable price.

4.1 Key Features

- Support SIP 2.0 (RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP, NTP, PPPoE, STUN, TFTP, etc.
- Powerful Digital Signal Processing (DSP) technology to ensure superior audio quality
- Advanced and patent pending adaptive jitter buffer control, packet delay and loss concealment technology
- Support various codecs including G.711 (PCM a-law and u-law), G.723.1 (5.3K/6.3K), G.729A, G.726 (32K) and iLBC
- Support standard voice features such as numeric Caller ID Display, Call Waiting, Hold, Transfer, Forward, 3-Way Conference, in-band and out-of-band DTMF, off hook autodial, auto answer.
- Support syslog, full duplex hands-free speakerphone, redial, call log, volume control, voice mail with indicator, downloadable ring tones.
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support standard encryption and authentication (DIGEST using MD5, MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Provide easy configuration through manual operation (phone keypad), Web interface or automated centralized configuration file via TFTP or HTTP.
- Support firmware upgrade via TFTP or HTTP.

4.2 Hardware Specifications

There are two models in the BudgeTone-100 family, namely:

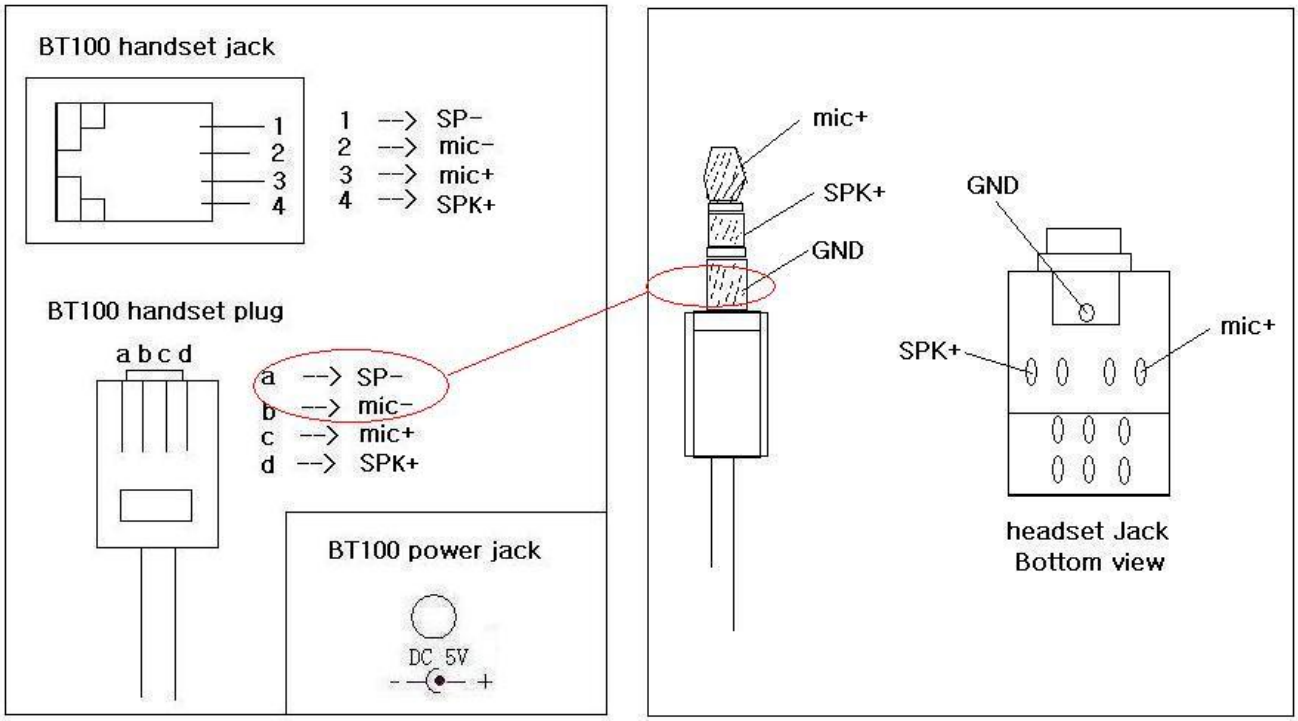
BudgeTone-101
BudgeTone-102 (As show in the below picture)

The ONLY difference between BT101 and BT102 is that the two RJ-45 ports of BT102 is actually a 10Base-T **mini-Hub** that allows user to share or sniffer the network using another data device like PC.

The table below describes the difference among these models.

Model	**BudgeTone-101**	**BudgeTone-102**
LAN interface	1xRJ45 10Base-T	2xRJ45 10Base-T
Phone Case	25-button keypad 12-digit caller ID LCD	25-button keypad 12-digit caller ID LCD
Headset Plug	3.5 mm	3.5 mm
Universal Switching Power Adaptor	Input: 100-240VAC Output: +5VDC, 400mA, UL certified	Same as left
Dimension	18cm (W) 22cm (D) 6.5cm (H)	Same as left
Weight	2 lbs (0.9 kg)	Same as left
Operating Temperature	32 - 104^{o}F 0 - 40^{o}C	Same as left
Humidity	10% - 95% (non-condensing)	Same as left
Compliance	FCC/CE/C-Tick	Same as left

The picture below shows the handset and headset connectors' wiring schema.

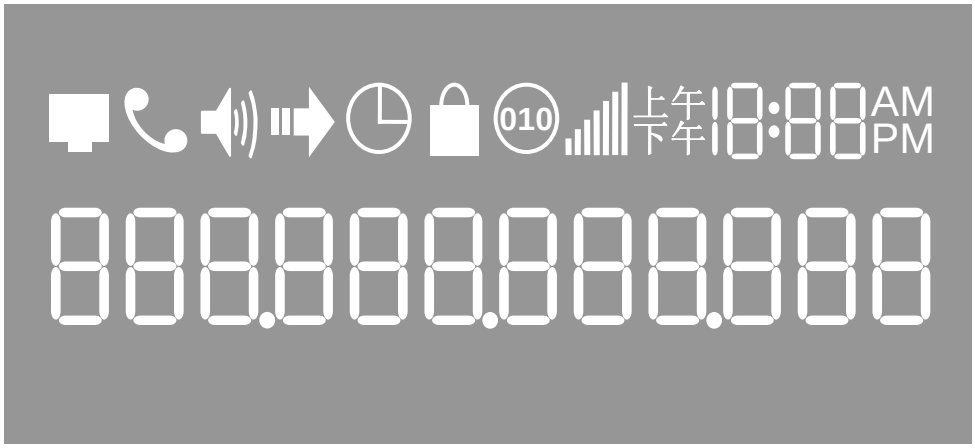


As show in the schema, the left side is pin assignment for a RJ11 interface headset; while the right side is showing a normal 3.5mm headset plug. A 3.5mm to 2.5mm plug converter is required if user want to user normal 2.5mm cell phone headset. The plug converter can be purchased from any electronics department store like Radio Shack.

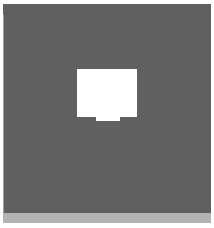
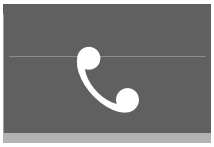
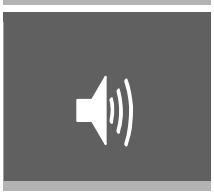
5 Basic Operations

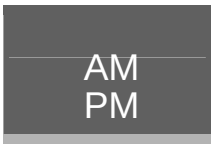
5.1 Get Familiar with LCD/LED

BudgeTone-100 phone has a numeric LCD of 64mmx24mm size with backlight. The new model (which is shipping now) has a small red LED status reminder. Here is the display when all segments illuminate:



When the phone is in the normal idle state, the backlight is off. Whenever an event (call) occurs, the backlight and the red LED will turn on automatically to bring the user's attention. In addition, if Voice Mail configured and there is a VM waiting, the backlight will be blinking and the red LED message light will light up to remind user there is a Voice Mail in the Voice Mail server.

Icon	LCD Icon Definitions
	Network Status Icon: FLASH in the case of Ethernet link failure or the phone is not registered properly. OFF if IP address or SIP server is not found ON if IP address and SIP server are located
	Phone Status Icon: OFF when the handset is on-hook ON when the handset is off-hook
	Speakerphone/Headset Status Icon: FLASH when phone rings OFF when the speakerphone/headset is off ON when the speakerphone/headset is on

	Handset and Speakerphone/Headset Volume Icons: 0-7 scales to adjust handset / speakerphone volume
	Real-time Clock: Synchronized to Internet time server Time zone configurable via web browser
	Call Logs: 01-10 for *CALLED* history (dialed number) 01-10 for *CALLERS* history (Incoming caller ID)
	Time Icon: AM for the morning PM for the afternoon
	IP Address Separator Icons:
	Numerical Numbers and Characters: 0 - 9 * = ⌞ # = ⌟ A, b, C, c, d, E, F, G, g, H, h, I, L, n, O, o, P, q, r, S, t, U, u, Y

5.2 Get Familiar with Keypad

BudgeTone-100 phone has a 25-button keypad.

Key Button	Key Button Definitions
0 - 9, *, #	Digit, star and pound keys are usually used to make phone calls
↓	1) Reduce handset, speakerphone/headset volume after off hook the phone via handset or speaker 2) Reduce ring tone volume when phone in IDLE and off hook to confirm the changed ring tone volume 3) Next menu item browsing when phone is in IDLE mode after MENU key pressed, off hook to interrupt and exit

↑	1) Increase handset, speakerphone/headset volume after off hook the phone via handset or speaker 2) Increase ring tone volume when phone in IDLE and off hook to confirm the changed ring tone volume 3) Previous menu item browsing when phone is in IDLE mode after MENU key pressed, off hook to interrupt and exit
MENU	Enter keypad MENU mode when phone is in IDLE mode. It is also the ENTER key once entering MENU
CALLED	After off hook, press to display the dialed numbers. When number displayed, press the SEND key can make call using that displayed number
CALLERS	After off hook, press to display the incoming Caller IDs. When number displayed, press the SEND key can make call using that displayed number
MESSAGE	Enter to retrieve voice mails from Voice Mail Portal or Server
HOLD	Temporarily hold the active call
TRANSFER	Transfer the active call to another party
CONFERENCE	Establish 3-way conferencing call
FLASH	Flash event to switch between two lines
MUTE/DEL	Mute an active call; or Delete a key entry, call log etc
SEND/(RE)DIAL	Dial a new number inputted or Redial the number last dialed. After entering the phone number, pressing this key would force a call to go out immediately before timeout
SPEAKERPHONE	Enter hands-free mode

5.3 Make Phone Calls

5.3.1 Make Calls using Numbers

There are three ways to make phone calls:

1. Pick up handset or press SPEAKERPHONE button, and then enter the phone numbers
2. Press the SEND button directly to redial the number last called.
   Once pressed, the last dialed number will be displayed on the LCD as the corresponding DTMF tones are played out and an outgoing call is sent.
3. Browse the CALLED/CALLER history and press the SEND/REDIAL button.
   Pick up the handset or press the speakerphone button, then press the "CALLED/CALLERS" button to browse thru the last 10 numbers dialed out. Once the desired number is identified and displayed on the LCD screen, press the SEND button and a new call to that displayed number will be sent out immediately.

Examples:

- To dial another extension on the same proxy, such as 1008, simply pick up handset or press speakerphone, dial 1008 and then press the "SEND" button.
- To dial a PSTN number such as 6266667890, you might need to enter in some prefix number followed by the phone number. Please check with your VoIP service provider to get the information. If you phone is assigned with a PSTN-like number such as 6265556789, most likely you just follow the rule to dial 16266667890 as if you were calling from a regular analog phone, followed by pressing the "SEND" button.

5.3.2 Make Calls using IP Address

Direct IP calling allows two parties, that is, a BudgeTone phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy. This kind of VoIP calls can be made between two parties if:

- Both BudgeTone phone and other VoIP Device(i.e., another IP Phone or BudgeTone SIP phone or other VoIP unit) have public IP addresses, or
- Both BudgeTone phone and other VoIP Device are on the same LAN using private or public IP addresses, or
- Both BudgeTone phone and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a direct IP to IP call, first off hook, then press *"MENU"* key, then enter a 12-digit target IP address to make the call. If port is not default 5060, destination ports can be specified by using "*4" (encoding for ":") followed by the port number.

Examples:

- If the target IP address is 192.168.0.10, the dialing convention is
  MENU_key 192 168 000 010
  followed by pressing the "SEND" key or wait for seconds in the No Key Entry Timeout.

- If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:
  MENU_key 192168001020*45062
  followed by pressing the "SEND" key wait for seconds in the No Key Entry Timeout.

5.3.3 Answer an Incoming Call

There are two ways to answer an incoming call:

1. Pick up the handset to answer the call normally using handset, or
2. Press the SPEAKERPHONE button to answer in speakerphone or headset mode

5.3.4 Handset Mode, Speakerphone/Headset Mode

Handset mode and Speakerphone/Headset mode cannot be enabled at the same time. Pressing the hook-switch or Speakerphone button would toggle the phone between these two modes.

5.3.5 Call Hold

While in conversation, pressing the "Hold" button will put the remote end on hold. Pressing the "Hold" button again will release the previously Hold state and resume the bi-directional media.

5.3.6 Call Waiting and Call Flashing

If call waiting feature is enabled, while the user is in a conversation, he will hear a special stutter tone if there is another incoming call. User then can press FLASH button to put the current call party on hold automatically and switch to the other call. Pressing flash button toggles between two active calls.

5.3.7 Call Transfer

Two transfer operations are supported.

5.3.7.1 Blind Transfer

User can transfer an active call to a third party without announcement.

User presses the "TRANSFER" button and if the other voice channel is available (i.e., there is no other active conversation besides the current one), user will hear a dial tone. User can then dial the third party's phone number followed by pressing SEND button.

NOTE:

- *"Enable Call Feature" has to be configured to "Yes" in web configuration page in order to make the features to work.*

A can hold on to the phone and wait for one of the three following behaviors:

- A quick confirmation tone (temporarily using the call waiting indication tone) follows by a dial tone. This indicates the transfer has been successful. At this point, the user can either hang up or make another call.
- A quick busy tone followed by a restored call (On supported platforms only). This means the transfer has failed due to the failed response sent from server and the phone will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
- Busy tone keeps playing. This means the phone has failed to receive the final response and decide to time out. Be advised that this does not indicate the transfer has been successful, nor does it indicate the transfer has failed.

5.3.7.2 Attended Transfer

User can transfer an active call to a third party with announcement.

User presses the "FLASH" button and hears a dial tone, then dial the third party's phone number followed by pressing "SEND" button. If the call is answered, press "TRANSFER" to complete the transfer operation and hand up, if the call is not answered, pressing "FLASH" button to resume the original call.

NOTE:

- *When Attended Transfer failed, if A hangs up, the BudgeTone phone will ring user A back again to remind A that B is still on the call. A can pick up the phone to restore conversation with B.*

5.3.8 Conference Call

BudgeTone 100 phone supports 3-way conference.

Assuming that call party A and B are in conversation. A wants to bring C in a conference:

1. A presses the "CONFERENCE" button to get a dial tone and put B on hold
2. A dials C's number then "SEND" key to make the call
3. If C answers the call, then A presses "CONFERENCE" button to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.

NOTE:

- *During the conference, if B or C drops the call, the remaining two parties can still talk. However, if A the conference initiator hangs up, all calls will be terminated.*

5.4 Call Features

Following table shows the available call features of BudgeTone 100 by using keypad star(*) code, if the VoIP service provider supports these call features in the server side:

Key	Call Features
*30	Block CallerID (for all-config change)
*31	Send CallerID (for all-config change)
*67	Block CallerID (per call base)
*82	Send CallerID (per call base)
*50	Disable Call Waiting (for all-config change)
*51	Enable Call Waiting (for all-config change)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial "*72" and get the dial tone. Then dial the forward number followed by pressing "SEND" button and hear dial tone again to confirm the forward, then hang up.
*73	Cancel Unconditional Call Forward To cancel "Unconditional Call Forward", dial "*73" and get the dial tone to confirm the cancel, then hang up.
*90	Busy Call Forward To use this feature, dial "*90" and get the dial tone. Then dial the forward number followed by pressing "SEND" button and hear dial tone again to confirm the forward, then hang up.
*91	Cancel Busy Call Forward To cancel "Unconditional Call Forward", dial "*91" and get the dial tone to confirm the cancel, then hang up
*92	Delayed Call Forward To use this feature, dial "*92" and get the dial tone. Then dial the forward number followed by pressing "SEND" button and hear dial tone again to confirm the forward, then hang up.
*93	Cancel Delayed Call Forward To cancel this Forward, dial "*93" and get the dial tone to confirm the cancel, then hang up
Flash or Hook Flash	When in conversation, this action will switch to the new incoming call if user hears the call waiting sound. When in conversation and no other incoming call, this action will switch to a new channel for a new call.

6 Configuration Guide

6.1 Configuration with Keypad

When the phone is IDLE or On Hook, press the *MENU* button to enter key pad menu state. When the phone goes off-hook or a call comes in, the phone automatically exits the key pad menu state and prepare for the call. It also exits the key pad menu state if left idle for 20 seconds.

Here are the key pad menu options supported:

Menu Item	Menu Functions
1	Display "**[1] dhcP On** " or "**[1] dhcP oFF**" for the current selection Press *MENU* key to enter edit mode Press '↓' or '↑' to toggle the selection Press *MENU* to save and exit Must recycle power to take effective!!!
2	Display "**[2] IP Addr** " Press *MENU* to display the current IP address Enter new IP address if DHCP is OFF Press '↓' or '↑' to exit Press *MENU* to (save and) exit Must recycle power to take effective!!!
3	Display "**[3] SubNet** " Press *MENU* to display the Subnet mask Enter new Subnet mask if DHCP is OFF Press '↓' or '↑' to exit Press *MENU* to (save and) exit Must recycle power to take effective!!!
4	Display "**[4] routEr** " Press *MENU* to display the Router/Gateway address Enter new Router/Gateway address if DHCP is OFF Press '↓' or '↑' to exit Press *MENU* to (save and) exit Must recycle power to take effective!!!

Menu Item	Menu Functions
5	Display "**[5] dnS**" Press *MENU* to display the DNS address Enter new DNS address if DHCP is OFF Press '↓' or '↑' to exit Press *MENU* to (save and) exit Must recycle power to take effective!!!
6	Display "**[6] tFtP**" Press *MENU* to display the TFTP address Enter new TFTP server IP address Press *MENU* to save Press '↓' or '↑' to exit
7	Display "**[7] G-711u 2**" Press *MENU* to select new codec Press '↓' or '↑' to browse a list of available codecs line 1 "**- G-711A 2**" 2 "**- G-722 2**" 3 "**- G-723 1**" 4 "**- G-726 2**" 5 "**- G-728 8**" 6 "**- G-729 2**" 7 "**- iLBC 1**" Press 1 to 9 to indicate number of frames per TX packet Press *MENU* to save and exit Must recycle power to take effective!!!
8	Display "**[8] SIP SP-1**" Reserve for future products.

Menu Item	Menu Functions
9	Display "**[9] codE rEL**" Press *Menu* to display the code releases Press '↓' or '↑' to browse line 1 "**b 2006-01-09**" – date: *boot* code 2 " **1. 0. 8. 9**" – version: *boot* code 3 "**P 2006-02-10**" – date: *phone* code 4 " **1. 0. 8. 16** – version: *phone* code 5 "**c 2005-03-05**" – date: *codec* 6 " **1. 0. 1. 0**" – version: *codec* 7 "**h 2006-02-10**" – date: *web* server 8 " **1. 0. 8. 16**" – version: *web* server 9 "1**r 2004-05-12**" – date: 1st *ring* tone 10 " **1. 0. 0. 0**" – version: *ring* tone 11 "2**r 2004-05-12**" – date: 2nd *ring* tone 12 " **1. 0. 0. 0**" – version: *ring* tone 13 "3**r 0000-00-00**" – date: 3rd *ring* tone 14 " **0. 0. 0. 0**" – version: *ring* tone (all zeroes means unavailable or unsupported) Press *MENU* to exit
10	Display "[**10] Phy Addr**" Press *MENU* to display the physical / MAC address Press '↓' or '↑' to exit
11	Display "[11] **ring 0**" Press *MENU* to hear the selected ring tone, press '↓' or '↑' to select the stored ring tones. Now only 3 are available, ring 0 (default), ring 1 and ring 2. ring 3 is unavailable or unsupported. Press *MENU* to select and exit
	Display " -- rESEt --", please be very **CAREFUL** here • Key in the physical / MAC address on back of the phone, Press MENU, phone will be reset to FACTORY DEFAULT setting, and all your setting will be erased. • Press MENU key without key in anything, phone will function the same as power cycle or reboot

Menu Item	Menu Functions
Others	When phone is powered on and time is displayed • Press '↓' or '↑', Display " ring [4] ", press '↓' or '↑' again to hear and adjust the ring tone volume, from 0 (off) to 7 (maximum), off and on hook to set • Press "SPEAKERPHONE" button, or off hook and pick up handset, press '↓' or '↑' to adjust the speakerphone/headset or handset volume

6.2 Configuration with Web Browser

BudgeTone-100 series IP phone has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the IP phone through a Web browser such as Microsoft's IE or Mozilla's Firefox..

6.2.1 Access the Web Configuration Menu

The BudgeTone-100 IP Phone Web Configuration Pages can be accessed by input the phones IP into browser's URL address field like:

http://Phone-IP-Address

where the ***Phone-IP-Address*** is the IP address of the phone. There are two ways to retrieve this IP address from the phone:

1) When the phone is *off-hook or in speakerphone* mode, simply press *MENU* button. *(This is most common way to get the IP address of the phone)*
2) When the phone is on-hook, press *MENU* button and then the browsing arrow keys to "**[2] IP Addr**", press *MENU* again.

NOTE:

- *To type IP address into browser to bring up the configuration pages, please strip out the leading "0" as the browser will parse in octet. e.g.: if the IP address is: 192.168.001.014, please type in: 192.168.1.14.*

Once the correct IP address of the phone is input into browser and "Enter" key pressed, the web log in page will come up like following:

The password is case sensitive with maximum length of 25 characters. The factory default password for End User is "123", for Administrator is "admin" respectively. Only administrator has the privilege to get access to "ADVANCED SETTINGS" configuration page.

NOTE:

- *If you cannot log into the configuration pages by using default password, please check with the VoIP service provider. Most likely, the service provider has already provisioned and automatically configured the device for you and has changed the default password.*

6.2.2 Configuration Menu

After input the correct password into the login screen, the embedded Web server of the IP phone will respond with the Configuration Pages screen, which is explained in details below.

- **Status Page:**

Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS

MAC Address:	00.0B.82.05.CA.91
IP Address:	192.168.1.101
Product Model:	BT100 REV 2.0
Software Version:	Program-- 1.0.8.16 Bootloader-- 1.0.8.9 HTML-- 1.0.8.16 VOC-- 1.0.1.0
System Up Time:	0 day(s) 18 hour(s) 1 minute(s)
Registered:	Yes
PPPoE Link Up:	disabled
NAT:	detected NAT type is open Internet
NAT Mapped IP:	0.0.0.0
NAT Mapped Port:	0
Total Inbound Calls:	7
Total Outbound Calls:	2
Total Missed Calls:	3
Total Call Time (in minutes):	19
Total SIP Message Sent:	428
Total SIP Message Received:	626
Total RTP Packet Sent:	38542
Total RTP Packet Received:	38514
Total RTP Packet Loss:	0

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MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting.
IP Address	This field shows IP address of BudgeTone 100.
Product Model	This field contains the product model info. Like BT100 (4M, discontinued legacy product) or BT100 REV2.0 (8M RAM)
Software Version	***Program:*** This is the main firmware release usually requested by support. ***Bootloader:*** This indicates bootloader version. ***HTML:*** This indicates the web interface version, normally not changed unless there is new parameter introduced in. ***VOC:*** This indicates the coder program version, normally not changed.
System Uptime	This shows system up time since last reboot.
Registered	This shows whether the unit is registered to VoIP service provider's server or proxy.
PPPoE Link Up	This shows whether the PPPoE is up if connected to DSL modem
NAT	This shows what kind of NAT the BudgeTone is behind if it is not in open interne, determined by STUN, usually is Full Cone, Port Restricted Cone etc., non-symmetric NAT if using general SOHO router
NAT Mapped IP	WAN side public IP of the NAT router if phone connected to LAN port of the router
Other Statistical Status of Phone	Self-explainable, see the page displayed.

- **Basic Settings:**

Grandstream Device Configuration

STATUS | BASIC SETTINGS | ADVANCED SETTINGS

End User Password: (purposely not displayed for security protection)

IP Address:

dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)

PPPoE account ID:

PPPoE password:

Preferred DNS server: 0 . 0 . 0 . 0

statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 2:

Time Zone: GMT-5:00 (US Eastern Time, New York)

Daylight Savings Time: No Yes (if set to Yes, display time will be 1 hour ahead of normal time)

Date Display Format:

Year-Month-Day

Month-Day-Year

Day-Month-Year

Update

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End user Password	This contains the password for end user to access the Web Configuration Menu. This field is case sensitive and maximum length is 25 characters. The default end user password is ***"123***". End user only has privilege to see the Status page and change parameters in Basic page. End user can NOT access to Advanced Settings pages and will get error message if try to.
IP Address	There are 2 modes under which the IP phone can operate: - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the IP phone will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to. . PPPoE account settings are configured here if the user is connecting the IP phone directly to the DSL/ADSL modem. Users can specify DNS server manually by entering DNS server's IP address. The IP Phone will attempt to establish a PPPoE session if PPPoE account is set. For most users, just leave them blank - If Static IP mode is selected, the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (mandatory), DNS Server 2 (optional) fields need to be configured. These fields are set to zero by default.
Time zone	Displayed date/time will be adjusted according to the specified time zone.
Day light savings time	Default NO. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
Date display format	This parameter controls the date display format.

- **Advanced Settings:**

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS

Admin Password:	(purposely not displayed for security protection)
SIP Server:	voip.domain.com (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	sbc.domain.com (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	6263441462 (the user part of an SIP address)
Authenticate ID:	6263441462 (can be identical to or different from **SIP User ID**)
Authenticate Password:	(purposely not displayed for security protection)
Name:	Tom Smith (optional, e.g., John Doe)
Advanced Options:	
Preferred Vocoder: (in listed order)	Choice1: G.729A/B choice2: G.723.1 choice3: iLBC choice4: PCMU choice5: PCMA choice6: G.729A/B choice7: iLBC choice8: G.726-32
G723 rate:	(•) 6.3kbps encoding rate ( ) 5.3kbps encoding rate
iLBC frame size:	( ) 20ms (•) 30ms
iLBC payload type:	97 (between 96 and 127, default is 97)
Silence Suppression:	(•) No ( ) Yes
Voice Frames per TX:	2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Layer 3 QoS:	48 (Diff-Serv or Precedence value)
Layer 2 QoS:	802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)
Allow incoming SIP messages from SIP proxy only:	(•) No ( ) Yes

Use DNS SRV:	○ No ○ Yes
User ID is phone number:	○ No ○ Yes
SIP Registration:	○ Yes ○ No
Unregister On Reboot:	○ Yes ○ No
Register Expiration:	3600 (in seconds. default 1 hour, max 45 days)
Early Dial:	○ No ○ Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	(this prefix string is added to each dialed number)
No Key Entry Timeout:	4 (in seconds, default is 4 seconds)
Use # as Dial Key:	○ No ○ Yes (if set to Yes, "#" will function as the Dial key)
local SIP port:	5060 (default 5060)
local RTP port:	5004 (1024-65535, default 5004)
Use random port:	○ No ○ Yes
NAT Traversal:	○ No ○ Yes, STUN server is: (URI or IP:port)
keep-alive interval:	20 (in seconds, default 20 seconds)
Use NAT IP	(if specified, this IP address is used in SIP/SDP message)
Proxy-Require:	(if specified, the content will appear in Proxy-Require header)
Voice Mail UserID:	8500 (User ID/extension for 3rd party voice mail system)
SUBSCRIBE for MWI:	○ No, do not send SUBSCRIBE for Message Waiting Indication ○ Yes, send periodical SUBSCRIBE for Message Waiting Indication
Auto Answer:	○ No ○ Yes
Offhook Auto-Dial:	(User ID/extension to dial automatically when offhook)
Enable Call Features:	○ No ○ Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
Disable Call-Waiting:	○ No ○ Yes
Send DTMF:	○ in-audio ○ via RTP (RFC2833) ○ via SIP INFO
DTMF Payload Type:	101

Send Flash Event:	○ No ○ Yes (Flash will be sent as a DTMF event if set to Yes)
Onhook Threshold:	800 ms
NTP Server:	ntp.nasa.gov (URI or IP address)
Default Ring Tone:	○ system ring tone ◉ custom ring tone 1, used if incoming caller ID is ○ custom ring tone 2, used if incoming caller ID is ○ custom ring tone 3, used if incoming caller ID is
Send Anonymous:	◉ No ○ Yes (caller ID will be blocked if set to Yes)
Anonymous Method:	◉ Use From Header ○ Use Privacy Header
Special Feature:	Standard
Syslog Server:	yoursyslog.com
Syslog Level:	DEBUG
Firmware Upgrade and Provisioning:	Upgrade Via ○ TFTP ◉ HTTP Firmware Server Path: fm.grandstream.com/gs Config Server Path: fm.grandstream.com/gs Firmware File Prefix: Firmware File Postfix: Config File Prefix: Config File Postfix: Automatic Upgrade: ○ No ◉ Yes, check for upgrade every 60 minutes (default 7 days) ◉ Always Check for New Firmware ○ Check New Firmware only when F/W pre/suffix changes
Firmware Key:	(in Hexadecimal Representation)
Authenticate Conf File:	◉ No ○ Yes (cfg file would be authenticated before acceptance if set to Yes)
Lock keypad update:	◉ No ○ Yes (configuration update via keypad is disabled if set to Yes)
Allow conf SIP Account in Basic Settings:	◉ No ○ Yes
Override MTU Size:	0

Update

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Admin Password	Administrator password. Only administrator can configure the "Advanced Settings" page. Password field is purposely blanked for security reason after clicking update and saved. The maximum password length is 25 characters.
SIP Server	IP address or FQDN domain name provided by VoIP service provider. e.g., the following are some valid examples: sip.my-voip-provider.com or sip:my-company-sip-server.com or 192.168.1.200:5066 (where 5066 is the port number different to default 5060)
Outbound Proxy	IP address or FQDN domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by IP phone for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy will provide solution for it.
SIP User ID	User account information, provided by VoIP service provider (ITSP), usually has the form of digits similar to phone number or actually a phone number
Authentication ID	ID used for authentication, usually same as SIP user ID, but could be different and decided by the ITSP
Authenticate Password	Account information, password for IP Phone to register to (SIP) servers of ITSP. Maximum length is 25 characters.
Preferred Vocoder	The BudgeTone IP phone supports up to 8 different codec types including G711-ulaw (PCMU), G711-alaw (PCMA), G723, G729A, G726-32 (ADPCM), G722, G728 and iLBC. A user can configure codecs in a preference list that will be included with the same preference order in SDP message. The first codec in this list can be entered by choosing the appropriate option in "Choice 1". Similarly, the last codec in this list can be entered by choosing the appropriate option in "Choice 8".
G723 Rate:	Encoding rate for G723 codec. By default, 6.3kbps rate is set.
iLBC frame size	iLBC packet frame size. Default is 20ms. For Asterisk IP-PBX, 30ms might need to be configured for compatibility
iLBC payload type	Payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to "Yes", when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled.

Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single Ethernet packet (be advised the max. size of Ethernet packet is 1500 byte (or 120k bit) so user should be aware that there IS a limit there). When setting this value, the user should be aware of the requested packet time (ptime, used in SDP message) as a result of configuring this parameter. This parameter is associated with the first codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first codec is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first codec chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the IP phone will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10 (x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively. Please be very careful when massage those parameters. By adjust this, user also get jitter buffer changed accordingly. BT-100 phone has patent dynamic jitter buffer handling algorithm. The jitter buffer range from 20 ~ 200 ms. Incorrect setting will affect voice quality so do not touch the parameter if not understand and most of the case the default value will work in GS products. Please refer to the Codec FAQ in our website for more technical details: http://www.grandstream.com/FAQ-Codec.pdf
Layer 3 QoS	This field defines the layer 3 QoS parameter, which can be used for IP Precedence or Diff-Serv or MPLS. Default value is 48.
Layer 2 QoS	Layer 2 QoS settings. Default setting is blank or "0" Other VLAN supported equipments like VLAN switch/router required if user wants to configure these settings.
Allow incoming SIP messages from SIP proxy only	If set to "Yes", the phone will ignore any SIP message that does not come from the IP address (Source IP in the IP header, the SIP server) that it is registered to. Default is No.
Use DNS SRV	Default is No. If set to Yes, the phone will use DNS SRV configured to lookup for the server
Use ID is phone number	If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request, which will be processed by supported SIP proxy. Default is No.
SIP registration	This parameter controls whether the BudgeTone phone needs to send REGISTER messages to the proxy server. The default setting is "Yes".

Unregister On Reboot	Default is No. If set to Yes, the phone will send "remove all register" request to the server ("*" in the contact header) to remove all previous bindings. If server does not support this it will cause some problems
Register Expiration	This parameter allows the user to specify the time frequency (in seconds) the IP phone will refresh its registration with the specified registrar (SIP Server). The default interval is 3600 seconds (or 1 hour). The maximum interval is 45 days.
Early Dial	Default setting is No. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response (like Asterisk). Otherwise, the call will most likely be rejected by the proxy (with a 404 Not Found error). Please note that this feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.
Dial Plan Prefix	Sets the prefix added to each dialed number. If configured, the prefix will be added to EVERY number input
No key Entry Timeout	Default is 4 seconds. User can short or extend that depends on digits dialed habit
Use # as Dial Key	This parameter allows the user to configure the "#" key to be used as the "SEND" key. Once set to "Yes", pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this "#" key is essentially equivalent to the "SEND" key. If set to "No", this # key will then be included as part of the dial string to be sent out.
Local SIP port	This parameter defines the local SIP port the IP phone will listen and transmit on. The default value is 5060.
Local RTP port	This parameter defines the local RTP-RTCP port pair the IP phone will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port_value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value is 5004.
Use Random port	Default No. If set to Yes, the device will pick randomly generated SIP and RTP ports. This is usually necessary and useful when multiple IP Phones are behind the same full cone NAT router.
NAT Traversal	Defines whether the NAT traversal mechanism is activated. It should be set to YES if the device is behind NAT router. If Outbound Proxy is NOT configured, STUN server needs to be set to activate STUN detection mechanism. Usually ITSP will provide these settings for device to work properly behind NAT/Firewall If this field is set to "Yes" without STUN server, then the device will periodically (every ***Keep-alive interval***) send a dummy UDP packet to the SIP server to pinhole the NAT in the router side.

Keep alive interval	Default is 20 seconds. The interval of sending dummy UDP packet to keep NAT "pin hole" open in the router side.
Use NAT IP	NAT IP address (WAN side) used in SIP/SDP message. Default is blank.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall. Required by some soft switch vendor like Nortel MCS.
Voice Mail User ID	User ID (extension or access number) of a 3rd party VoiceMail system where the user may have an account. By defining it, user presses the "MESSAGE" button on the phone, an INVITE message will send to that ID/number to allow the user to retrieve VM.
Subscribe for MWI	Default is No. When set to Yes, a SUBSCRIBE for Message Waiting Indication will be sent periodically to server. BT-100 support both synchronize and non-synchronized SUBSCRIBE SIP message.
Auto Answer	Default is No. When set to Yes, the phone will automatically pick up the call after a short beep and turn on the speaker.
Offhook Auto-Dial	This parameter allows the user to configure a User ID or extension number to be automatically dialed upon off hook (like hot line). Please note that only the user part of a SIP address needs to be entered here. The phone will automatically append the "@" and the host portion of the corresponding SIP address.
Enable call features	Default is Yes. Advance call features or feature codes functions (Star code, see Section 5.4 of this manual) are supported locally
Disable Call Waiting	Default is No. User can use * code to use this feature per call basis.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digits. There are 3 modes supported: in audio, which means DTMF is combined in in-band audio signal (not very reliable with low-bit-rate codec); via RTP (RFC2833); or via SIP INFO. Make sure this setting is synchronized with SIP server otherwise will not work.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833 Default is 101
Send Flash Event	Default is NO. If set to Yes, flash will be sent as DTMF event therefore the FLASH key to switch calls will NOT work.
Onhook Threshold	Time period when cradle pressed (Hook Flash) to simulate FLASH. Adjust this time value will prevent user on hook carelessly to activate Flash/Hold and cause the phone ring back. Default is 800ms.
NTP server	URI or IP address of the NTP (Network Time Protocol) server, which will be used by the phone to synchronize the date and time.

Default Ring Tone	• By default "System Ring Tone" is selected. All calls received that do not match any "incoming caller ID" number will generate the selected ring tone - system, 1, 2 or 3. • The user can setup up one number on each of the ring tones 1, 2, and 3. When a call is received from one of these numbers, the respective ring tone will be generated. • If the ring tone does not exist, the system default ring tone will be played. For BT-100, ring 3 is not supported.
Send Anonymous	If this parameter is set to "Yes", user ID will be sent as anonymous, essentially block the Caller ID from displaying.
Anonymous Method	This is decided by supported SIP server/proxy. Either "Use From Header" or "Use Privacy Header" depends on how SIP sever/proxy process the SIP header.
Special Feature	Default is **Standard**. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Sonus, Broadsoft, etc. End user please do not touch this parameter.
Syslog Server	The IP address or URL of System log server, especially useful for ITSP (Internet Telephone Service Provider). End user don't encourage to touch this part

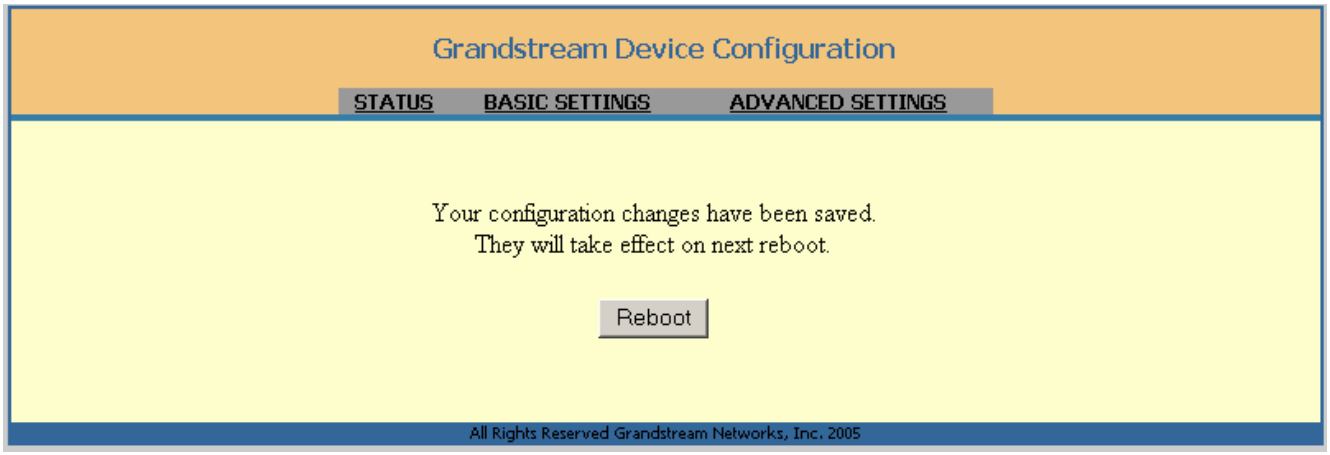
Syslog level	Select the level for the phone to report the log messages. Default is NONE. Useful for ITSP. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up

Firmware Upgrade and Provisioning	**Firmware upgrade and Provisioning can be done via either TFTP or HTTP. This is mutual exclusive choice.** **Default if HTTP.** Via TFTP: *(default is pointing to Grandstream Firmware TFTP server)* **The TFTP server needs to be configured. If it is non-zero or not blank, the IP phone will attempt to retrieve new configuration file or new firmware from the specified TFTP server at boot time. It will make up to 3 attempts before timeout and then it will start the boot process using the existing code image in the Flash memory.** ***Be very careful when the TFTP is in progress (the phone's LCD backlight will be ON). Do NOT interrupt the process (especially the power supply) otherwise might make the phone broken.*** ***Be patient as in some network environment this process will take up to 15 minutes.*** Via HTTP: *(default is pointing to Grandstream Firmware HTTP server)* **The URI for the web server used for firmware upgrade and configuration. For example,** **http://provisioning.mycompany.com:6688/Grandstream/1.0.8.16** **Here ":6688" is the specific TCP port that the HTTP server is listening to, it can be omitted if using default port 80.**
Firmware Server Path	IP address or domain name (URI) of firmware server. Default to Grandstream's firmware server
Config Server Path	IP address or domain name (URI) of configuration server. Default to Grandstream's server. It can be different with Firmware server.
Firmware File Prefix	Default blank. If configured, the phone will request the firmware file with the prefix. Useful for provisioning of ITSPs. End user should keep it blank.
Firmware File Postfix	Default is blank. Similar to Prefix. End user should keep it blank.
Config File Prefix	Default is blank. The phone will request configure file prefix is configured. Used by ITSP for provisioning. End user should keep it blank.
Config File Postfix	Default is blank. End user should keep it blank.

Automatic Upgrade	Default is Yes. Choose Yes to enable automatic HTTP upgrade and provisioning. In "Check for upgrade every" field, enter the number of minutes to enable the phone to check the HTTP server for firmware upgrade or configuration changes in the defined period of time. When set to No, the phone will only do HTTP upgrade and configuration check once at boot up. Used by ITSP. End user should NOT touch these parameters.
Always Check for New Firmware	The phone will check for new firmware once reboot and also when the configured automatic upgrade time period comes
Check New Firmware only when F/W pre/suffix changes	The phone will check for new firmware once reboot and also when the configured automatic upgrade time period comes, upon detected the pre/suffix changes. Otherwise will quit the planed process.
Firmware Key	32 digits in Hexadecimal. Once configured, the firmware will ONLY be changed if the key is matched. This will lock the unit and firmware by ITSP. Useful for ITSP to encrypt firmware. End user should keep it blank.
Authenticate Conf File	Default No. Useful by ITSPs. End user should use default setting. Once configured, only authenticated configuration file will be used.
Lock keypad update	Default is No. The configuration update via keypad is disabled if set to Yes. User can not change anything via key pad and only WebUI and configuration file can be used to change setting if set to Yes.
Allow conf SIP Account in Basic Settings	Default NO. If set to Yes, User ID, Authentication ID and Password can be configured in basic settings. This parameter is decided by related ITSP.
Override MTU Size	Override the MTU size to meet some special network settings. Default is "0", means no override.

6.2.3 Saving the Configuration Changes

Once a change is made, the user should press the "Update" button in the Configuration Menu. The IP phone will then display the following screen to confirm that the changes have been saved.



User is recommended to reboot or power cycle the IP phone after all the changes are made so that those changes can take effect.

6.2.4 Rebooting the Phone from Remote

The administrator of the phone can remotely reboot the phone by pressing the "Reboot" button at the bottom of the configuration menu. Once done, the following screen will be displayed to indicate that rebooting is underway.

Grandstream Device Configuration
The device is rebooting now...
You may relogin by clicking on the link below in 30 seconds.
Click to relogin
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At this point, the user can relogin to the phone after waiting for about 30 seconds.

6.3 Configuration through a Central Server

Grandstream IP phone can be automatically configured from a central provisioning system.

When BudgeTone phone boots up, it will send TFTP or HTTP request to download configuration file. The name of the configuration file is "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the BudgeTone.

The configuration files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of BudgeTone phone can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of BudgeTone. GAPS (Grandstream Automated Provisioning System) uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual BudgeTone phone for firmware upgrade, configuration parameters change or remote reboot, etc.

Grandstream provide GAPS (Grandstream Automated Provisioning System) service to VoIP service providers. It could be either simple redirection or with certain special provisioning settings. Initially upon booting up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer's TFTP or http server for further provisioning. Grandstream also provide GAPSLite software package which contains our NAT friendly TFTP server and a configuration tool to facilitate the task of generating device configuration files.

The GAPSLite configuration tool is now free to end users. The tool and configuration templates can be downloaded from http://www.grandstream.com/DOWNLOAD/Configuration_Tool/.

For details on how GAPS works, please refer to the documentation of GAPS product.

7 Software Upgrade

Software (or firmware) upgrade can be done either via TFTP or HTTP. The corresponding configuration settings are on the configuration page. End users should NOT touch the configuration settings that are useful for ITSPs.

7.1 Upgrade through HTTP

To upgrade firmware via HTTP, the field *"Firmware Upgrade and Provisioning: Upgrade Via"* needs to be set to HTTP. The *"Firmware Server Path"* should be set to where the firmware files are located.

For example, user can fill the following URL in the Firmware Server Path:

firmware.mycompany.com:6688/Grandstream/1.0.8.16

where firmware.mycompany.com is the FQDN of the HTTP server. It can also be in IP address format. ":6688" is the TCP port the HTTP server listening to, default http server listens to port 80. "/Grandstream/1.0.8.16" is the RELATIVE directory to the root dir on HTTP web server.

7.2 Upgrade through TFTP

To upgrade firmware via TFTP, the field *"Firmware Upgrade and Provisioning: Upgrade Via"* needs to be set to TFTP.

TFTP server can be configured in either IP address format or FQDN.

There are two ways to set up the TFTP server in IP format to upgrade the firmware, namely through the Key Pad MENU or via the BudgeTone phone's Web configuration interface. To configure the TFTP server via Key Pad, please refer to Section 6.1 of this user manual. If TFTP server is in FQDN format, it must be set via web configuration interface.

To configure the TFTP server via the Web configuration interface, open up your browser to input the IP address of the BudgeTone phone. Enter the admin password to enter the configuration screen. From there, enter the TFTP server address or URL in the "Firmware Server Path" field near the bottom of the configuration screen. Once the "Firmware Server Path" is set, user needs to update the change by clicking the "Update" button. Then "Reboot" or power cycle the phone, the firmware files will be fetched upon booting up.

If the configured updating server is found and a new code image is available, the BudgeTone phone will attempt to retrieve the new image files by downloading them into the BudgeTone phone's SRAM. During this stage, the BudgeTone phone's LED/LCD will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files

available for upgrade, or checksum test fails, etc), the BudgeTone phone will stop the TFTP process and simply boot using the existing code image in the flash.

Firmware upgrading may take as long as 1 to 20 minutes over Internet, or just 20+ seconds if it is performed on a LAN. It is generally recommended to conduct firmware upgrade in a controlled LAN environment if possible.

For users who do not have local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Services section of Grandstream's Web site to obtain this TFTP server IP address:

Alternatively, user can download and install free TFTP or HTTP server in his LAN to do firmware upgrading.

A free Windows version TFTP server can be downloaded from:
http://support.solarwinds.net/updates/New-customerFree.cfm.
Our latest official release can be downloaded from:
http://www.grandstream.com/y-firmware.htm
Unzip the file and put all of them under the root directory of the TFTP server. Put the PC running the TFTP server and the BT-100 phone in the same LAN segment. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade. Start the TFTP server, in the phone's web configuration page, configure the Firmware Server Path with the IP address of the PC, update the change and reboot the unit.

User can also choose to download the free HTTP server from http://httpd.apache.org/ or just use Microsoft IIS web server

NOTE:

- *When BudgeTone phone boots up, it will send TFTP or HTTP request to download configuration file "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the BudgeTone phone. This file is for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.*

  ```
  TFTP Error from [IP ADRESS] requesting cfg000b82023dd4 : File does not exist
  Configuration File Download
  ```

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP. "Config Server Path" is the TFTP or HTTP server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The "Config Server Path" can be same or different from the "Firmware Server Path".

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with "Admin Password" in the ADVANCED SETTINGS page. For a

detailed parameter list or P values, please refer to the corresponding firmware release configuration template of the related products.

When Grandstream Device boots up or reboots, it will issue request for configuration file named "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., cfg000b820102ab". The configuration file name should be in lower cases.

7.3 Firmware and Configuration File Prefix and Postfix

Starting from firmware version 1.0.8.16 for BT-100 phone Rev 2.0, adding prefix and postfix for both firmware and configuration file is supported.

Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it the possible to store ALL of the firmware with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Therefore multiple configuration files for the same device can be stored in one directory.

In addition, when the field "Check New Firmware only when F/W pre/suffix changes" is set to "Yes", the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

7.4 Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set to "Yes", Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check with either Firmware Server or Config Server, whenever they are defined. This allows the device periodically check if there are any new changes need to be taken on a scheduled time. By defining different intervals in P193 for different devices, Server Provider can spread the Firmware or Configuration File download in minutes to reduce the Firmware or Provisioning Server load at any given time.

8 Restore Factory Default Setting

WARNING !!!

Restore the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

Please disconnect network cable and power cycle the unit before trying to reset the unit to factory default. The steps are as follows:

Step 1:
Find the MAC address of the device. It is a 12 digits HEX number located on the bottom of the unit.

Step 2:
Encode the MAC address. Please use the following mapping:

0-9: 0-9
A: 22 (when pressed 2 twice, the "A" letter will show on the LCD)
B: 222
C: 2222
D: 33
E: 333
F: 3333

For example, if the MAC address is 000b8200e395, it should be encoded as "0002228200333395".

Step 3:
To perform factory reset:

a. Press the **MENU** button for Key Pad Menu options.
b. Press the Up or Down button to see "reset".
c. Enter the encoded MAC address.
d. Press the **MENU** button again
e. Wait for phone reboot and the LCD backlight finish flashing.

9 Glossary of Terms

ADSL

Asymmetric Digital Subscriber Line: Modems attached to twisted pair copper wiring that transmit from 1.5 Mbps to 9 Mbps downstream (to the subscriber) and from 16 kbps to 800 kbps upstream, depending on line distance.

AGC

Automatic Gain Control, is an electronic system found in many types of devices. Its purpose is to control the gain of a system in order to maintain some measure of performance over a changing range of real world conditions.

ARP

Address Resolution Protocol is a protocol used by the Internet Protocol (IP) [RFC826], pecifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when IPv4 is used over Ethernet

ATA

Analogue Telephone Adapter. Covert analogue telephone to be used in data network for VoIP, like Grandstream HT series products.

CODEC

Abbreviation for Coder-Decoder. It's an analog-to-digital (A/D) and digital-to-analog (D/A) converter for translating the signals from the outside world to digital, and back again.

CNG

Comfort Noise Generator, geneate artificial background noise used in radio and wireless communications to fill the silent time in a transmission resulting from voice activity detection.

DATAGRAM

A data packet carrying its own address information so it can be independently routed from its source to the destination computer

DECIMATE

To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while subsampling.

DECT

Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan-European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area

Networks, and wireless local loop. The DECT Common Interface radio standard is a multicarrier time division multiple access, time division duplex (MC-TDMA-TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each divided into 24 time slots of 10ms, and twelve full-duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multiframe (16 frames). Voice signals are digitally encoded into a 32 kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS

Short for ***D**omain **N**ame **S**ystem* (or ***S**ervice* or ***S**erver*), an Internet service that translates *domain names* into IP addresses

DID

Direct Inward Dialing

Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto-attendant.

DSP

Digital Signal Processing. Using computers to process signals such as sound, video, and other analog signals which have been converted to digital form.

Digital Signal Processor. A specialized CPU used for digital signal processing.

Grandstream products all have DSP chips built inside.

DTMF

Dual Tone Multi Frequency

The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, * and A-F) although most terminals support only 12 of them (0-9, * and #).

FQDN

Fully Qualified Domain Name

A FQDN consists of a host and domain name, including top-level domain. For example, www.grandstream.com is a fully qualified domain name. www is the host, grandstream is the second-level domain, and.com is the top level domain.

FXO

Foreign eXchange Office

An FXO device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like AT&T. They should also operate the same way when connected to an FXS interface.

An FXO interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.

FXO is complimentary to FXS (and the PSTN).

FXS

Foreign eXchange Station

An FXS device has hardware to generate the ring signal to the FXO extension (usually an analog phone).

An FXS device will allow any FXO device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.

The FXS Interface connects to FXO devices (by an FXO interface, of course).

DHCP

The *Dynamic Host Configuration Protocol* (DHCP) is an Internet protocol for automating the configuration of computers that use TCP/IP. DHCP can be used to automatically assign IP addresses, to deliver TCP/IP stack configuration parameters such as the subnet mask and default router, and to provide other configuration information such as the addresses for printer, time and news servers.

ECHO CANCELLATION

Echo Cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. In addition to improving quality, this process improves bandwidth savings achieved through silence suppression by preventing echo from traveling across a network.

There are two types of echo of relevance in telephony: acoustic echo and hybrid echo. Speech compression techniques and digital processing delay often contribute to echo generation in telephone networks.

H.323

A suite of standards for multimedia conferences on traditional packet-switched networks.

HTTP

Hyper Text Transfer Protocol; the World Wide Web protocol that performs the request and retrieve functions of a server

IP

Internet Protocol. A packet-based protocol for delivering data across networks**.**

IP-PBX

IP-based Private Branch Exchange

IP Telephony

(Internet Protocol telephony, also known as Voice over IP Telephony) A general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). The basic steps involved in originating an IP Telephony call are conversion of the analog voice signal to digital format and compression/translation of the signal into Internet protocol (IP) packets for transmission over the Internet or other packet-switched networks; the process is reversed at the receiving end. The terms IP Telephony and Internet Telephony are often used to mean the same; however, they are not 100 per cent interchangeable, since Internet is only a subcase of packet-switched networks. For users who have free or fixed-price Internet access, IP Telephony software essentially provides free telephone calls anywhere in the world. However, the challenge of IP Telephony is maintaining the quality of service expected by subscribers. Session border controllers resolve this issue by providing quality assurance comparable to legacy telephone systems.

IVR

IVR is a software application that accepts a combination of voice telephone input and touch-tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e-mail and perhaps other media.

MTU

A Maximum Transmission Unit (MTU) is the largest size packet or frame, specified in octets (eight-bit bytes), that can be sent in a packet- or frame-based network such as the Internet. The maximum for Ethernet is 1500 byte.

NAT

Network Address Translation

NTP

Network Time Protocol, a protocol to exchange and synchronize time over networks
The port used is UDP 123
Grandstream products using NTP to get time from Internet

OBP/SBC

Outbound Proxy or another name Session Border Controller. A device used in VoIP networks. OBP/SBCs are put into the signaling and media path between calling and called party. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called party. The effect of this behaviour is that not only the signaling traffic, but also the media traffic (voice, video etc) crosses the OBP/SBC. Without an OBP/SBC, the media traffic travels directly between the VoIP phones. Private OBP/SBCs are used along with firewalls to enable VoIP calls to and from a protected enterprise network. Public VoIP service providers use OBP/SBCs to allow the use of VoIP protocols from private networks with internet connections using NAT.

PPPoE

Point-to-Point Protocol over Ethernet, is a network protocol for encapsulating PPP frames in Ethernet frames. It is used mainly with cable modem and DSL services.

PSTN

Public Switched Telephone Network

i.e. the phone service we use for every ordinary phone call, or called POT (Plain Old Telephone), or circuit switched network.

RTCP

Real-time Transport Control Protocol, defined in RFC 3550, a sister protocol of the Real-time Transport Protocol (RTP), It partners RTP in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session. The primary function of RTCP is to provide feedback on the quality of service being provided by RTP.

RTP

Real-time Transport Protocol defines a standardized packet format for delivering audio and video over the Internet. It was developed by the Audio-Video Transport Working Group of the IETF and first published in 1996 as RFC 1889

SDP

Session Description Protocol, is a format for describing streaming media initialization parameters. It has been published by the IETF as RFC 2327.

SIP

Session Initiation Protocol, An IP telephony signaling protocol developed by the IETF (RFC3261). SIP is a text-based protocol suitable for integrated voice-data applications. SIP is designed for voice transmission and uses fewer resources and is considerably less complex than H.323.
All Grandstream products are SIP based

STUN

Simple Traversal of UDP over NATs, is a network protocol allowing clients behind NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. The protocol is defined in RFC 3489. STUN will usually work good with non-symmetric NAT routers.

TCP

Transmission Control Protocol, is one of the core protocols of the Internet protocol suite. Using TCP, applications on networked hosts can create *connections* to one another, over which they can exchange data or packets. The protocol guarantees reliable and in-order delivery of sender to receiver data.

TFTP

Trivial File Transfer Protocol, is a very simple file transfer protocol, with the functionality of a very basic form of FTP; It uses UDP (port 69) as its transport protocol.

UDP

User Datagram Protocol (UDP) is one of the core protocols of the Internet protocol suite. Using UDP, programs on networked computers can send short messages known as *datagrams* to one another. UDP does not provide the reliability and ordering guarantees that TCP does; datagrams may arrive out of order or go missing without notice. However, as a result, UDP is faster and more efficient for many lightweight or time-sensitive purposes.

VAD

Voice Activity Detection or Voice Activity Detector is an algorithm used in speech processing wherein, the presence or absence of human speech is detected from the audio samples.

VLAN

A virtual LAN, known as a VLAN, is a logically-independent network. Several VLANs can co-exist on a single physical switch. It is usually refer to the IEEE 802.1Q tagging protocol.

VoIP

Voice over IP

VoIP encompasses many protocols. All the protocols do some form of signalling of call capabilities and transport of voice data from one point to another. e.g: SIP, H.323, etc.